cover the extension fees.

Specification Amendments

Pursuant to 37 C.F.R. 1.125, applicant is providing a clean version and marked up version of substitute specification pages 16-18. Applicant is submitting substitute specification pages because, upon review, it has been determined that symbols, which were in the electronic version of the application, failed to properly print. While the substance of the information and equations represented by the symbols clearly communicates the nature and scope of the invention, applicant believes that, for purposes of clarity, substitute specification pages with the symbols should be provided.

SUBSTITUTE SPECIFICATION CLEAN VERSION

a new signal s 540, referred to herein as the echo response 540. The echo response 540 is illustrated here as a signal s 540 corresponding to the following equation:

$$s=h*x$$

where h is the impulse response of the echo characteristics.

Also being communicated through the transmit line 530 is a near-end signal v 545, communicated from a near-end source 550. The input signal v 545 combines with the echo response s 540 to generate a combined signal y 555. Therefore, the signal sent from the near-end source 550 to the far-end receiver 580, absent echo cancellation, is the signal y 555, which is the sum of the near-end signal v 545 and the echo response s 540.

To reduce and/or eliminate the echo response component s 540 from the signal y 555, the a typical system uses an echo canceller 560 having a filter 565 that is capable of applying an impulse response $\hat{h}$, which is an estimate of the actual impulse echo response h experienced by the far-end signal x 510 as it engages the cross-coupling path 525. As such, a further signal $\hat{s}$ 570 representing an estimate of echo response s 540 is generated by the echo canceller 560 in accordance with the following equation:

$$\hat{s} = \hat{h} * x$$

The echo canceller 560 subtracts the echo estimate signal $\hat{s}$ 570 from the signal y 555 to generate a signal e 575 that is returned to the far-end receiver 580. The signal e 575 thus corresponds to the following equation:

$$e = s+v- \hat{s} \approx v$$

The signal returned to the far end receiver 580 is therefore dominated by the signal v of the near-end source 550. To the extent the impulse response $\hat{h}$ more closely correlates to the actual echo impulse response h, then $\hat{s}$ 570 more closely approximates s 540, resulting in the minimization of the magnitude of the echo signal component s 540 on the signal e.

An adaptive filter 565 is used to generate the echo signal component $\hat{s}$ 570. In its simplest form, the adaptive filter 565 generates an echo estimate, i.e. $\hat{s}$ 570, by obtaining individual samples of the far-end signal x 510 on a receive path 513, convolving the samples with an impulse response model of the system, i.e. $\hat{h}$, and then subtracting, at the appropriate time, the resulting echo estimate, $\hat{s}$ 570, from the received signal y 555 on the transmit channel

530. The conventional adaptive filter is a FIR filter using a LMS method for achieving tap convergence.

The novel adaptive filter method and system presented herein can be used to improve the calculation of the echo impulse response $\hat{h}$ by, among other things, reducing the computational complexity and memory requirements of the tap calculation conducted within the adaptive filter. Shown in Figure 6, an embodiment of the novel filter 665 of the present invention is used to generate the echo signal component $\hat{s}$ 670. After having achieved convergence on a FIR filter and converted the filter to an IIR filter, in accordance with the previously described methodology, the adaptive filter 665 generates an echo estimate, i.e. $\hat{s}$ 670, by obtaining individual samples of the far-end signal x 610 on a receive path 613, convolving the samples with the calculated taps, and then subtracting, at the appropriate time, the resulting echo estimate, $\hat{s}$ 670, from the received signal y 655 on the transmit channel 630. On going adaptation of the filter occurs by the adjustment of the zeroes of the IIR filter, represented by the arrow 690 extending through element 680 (where $N_{iir}(z)$ denotes the numerator portion of the IIR filter), and not by updating the denominator 675. To match the delay incurred due to conversion of the FIR filter into an IIR filter, a delay where D-1 is a specific value of delay is applied. The signal $\hat{s}$ 670 is produced as a function of the transfer function denoted by z^{-D} 685.

As discussed above, to avoid degrading system performance through inaccuracies in detecting the start and/or end of the response, it is preferred in echo cancellation applications to truncate the converged FIR filter, take a first set of taps, K, from the truncated FIR filter, referred to as h_{fir}, take the last N-K taps of the truncated FIR filter, referred to as h_{iir}, and convert h_{iir} to an IIR model, where K is preferably at or around 10. The truncated FIR filter, h_{fir}, together with the IIR filter are then used, in combination, to track the system response and filter data.

Referring now to Figure 7, a second embodiment of the novel filter 765 of the present invention is used to generate the echo signal component $\hat{s}$ 770. After having achieved convergence on a FIR filter, dividing the filter taps into an initial K tap and a subsequent N-K coefficients, and converting a portion of the FIR filter to an IIR filter corresponding to the N-K taps, in accordance with the previously described methodology, the adaptive filter 765 generates an echo estimate, i.e. $\hat{s}$ 770, by utilizing both the truncated FIR filter 740, comprising $H_{fir}(z)$ 748,

and the IIR filter 745. On going adaptation of the IIR filter 745 occurs by the adjustment of the zeroes of the IIR filter, represented by the arrow 790 extending through element 780 (where $N_{iir}(z)$ denotes the numerator portion of the IIR filter), and not by updating the denominator 785. To match the delay incurred due to conversion of the FIR filter into an IIR filter, a delay where D1-1 is a specific value of delay is applied. The signal $\hat{s}$ 770 is produced as a function of the transfer functions denoted by z^{-D1} 743 and z^{-D2} 747.

Operationally, the novel echo cancellation application has achieved superior performance results in the form of computational savings and decreased filter length. An FIR filter of length N=512 was chosen, converged, and truncated according to the description provided above. The FIR filter was converted to an IIR filter using a pole-zero filter model of p=50 and q=50. To evaluate the ability of the echo cancellation system to adapt to changes over time, two actual hybrid responses, shown as 805, 830 in Figure 8 and as 905, 930 in Figure9, were generated in a PSTN due to an impedance mismatch of a four-wire to two-wire converter and recorded at two different times with an interval of 30 minutes. The two impulse responses 805/905, 830/930 demonstrate that, over time, changes do occur in an impulse response requiring an echo cancellation system, and more specifically, an adaptive filter, to adjust over time. Although on the order of 10^{-3} 940, the differences are sufficient to cause a converged system to generate, over time, a measured error that is unacceptable.

Without an adaptive filter, as shown in Figures 10 and 11, when the impulse response is switched close to 175,000 samples, the error 1050, 1150 increases significantly and to unacceptable levels. The error is measured in amplitude in Figure 10 and in decibels in Figure 11. Conversely, when the echo cancellation system employs one embodiment of the novel adaptive filter system claimed herein, the error 1250, 1350 shows an increase due to a shift in the impulse response but, unlike with a no-adaptive filter case, is at or below acceptable levels. If the filter order were increased, the error level would be further decreased, although memory requirements and computational resource needs would increase.

The present adaptive filter method and system can be employed in numerous applications employing adaptive processes in conjunction with convolutional coding. Accordingly, another embodiment of the present invention includes a novel method and system for channel equalization. Equalizers are a class of communication system devices used to compensate for distortion experienced in communication channels. Fixed equalizers have the average electrical